

DECIBELS
AUDIO AMPLIFICATION

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Decibels—Audio Amplification

AUDIO AMPLIFICATION MEASUREMENTS

Speech and music consist of many fundamental frequencies and with each are associated many harmonics and overtones. The intensities of the fundamental frequencies and their harmonics, and the distribution of harmonic and overtone energy differ widely for various persons and instruments, even though each may say the same word or sound the same note. It is this radical difference which makes possible identification of a voice or an instrument and determines the *quality* of the sound.

No matter how many frequencies are involved in a musical program, or in fact any sound program, the electrical amplifying and transmitting system must not alter a single one of these frequencies in the least. At any rate, this is the goal of all audio amplifier designers—amplification of the whole audible frequency range without altering the intensity of any frequency or frequencies with respect to the other frequencies and the original, both at low and high frequencies.

The radio broadcasting station sends out radio signals modulated with audio signals. These are picked up by a receiver, demodulated and fed into an audio system with a reproducer in its output. In public address systems, we have the phonograph disc and the phono-pickup or a microphone, by means of which audio signals are translated into electrical impulses and fed into an audio system. In sound pictures there is the film, or the phonograph record, which supplies the audio signals.

In every case there is an audio amplifier and wherever there is audio amplification we are vitally interested in the question “How closely does the output of the audio system resemble the original?”, regardless of whether the original was good or bad musically. We want to know whether anything has been added to it or taken away from it or whether an interfering signal has caused the reproduced sound to differ from the original sound.

The human ear ordinarily responds to frequencies between 20 and 15,000 cycles—and we can differentiate between low and high tones, and the musical sounds made by various instru-

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ments. Most of us can recognize the tones of a wind instrument (a bass horn, a French horn, a piccolo); a percussion instrument (drum); a string instrument (bass violin, cello, violin); and the human voice (soprano, alto, tenor or bass). Voice tones consist of fundamental frequencies varying from 50 to 1,500 cycles per second with overtone frequencies as high as 6,000 cycles. Sounds produced on musical instruments have fundamental frequencies ranging from 30 to 5,000 cycles while their harmonics and overtones include frequencies as high as 16,000 cycles.

A careful study of the human voice has shown that for voice reproduction, an audio amplifier should have a fidelity range of 250 to 2,500 cycles in order to amplify all the needed harmonics. Most of the energy of the human voice is carried in frequencies below 1,000 cycles per second but what we consider good tone quality is possible only with the inclusion of the frequencies above 1,000 cycles. In order that music may be reproduced with satisfactory and complete naturalness, the range to which the amplifier must respond uniformly is considered as being between 30 and 8,000 cycles. For radio purposes, perfect transcription in the audio system between 30 and 5,000 cycles is considered satisfactory, assuming that the radio frequency section of the receiving system does not discriminate by cutting side bands.

A modern transmitter is usually considered as almost perfect if it sends out programs uniformly over frequencies between 30 and 10,000 cycles with less than 10 per cent variation in signal strength, using 1,000 cycles as the standard audio frequency for purposes of comparison. Station WEAF of New York City has been considered as an almost perfect station for some time because it has been accomplishing this. Not many stations show the same characteristics and in smaller ones, transmission of audio frequencies below 170 cycles may be quite poor.

However, in our discussion of audio amplification, we must assume that the broadcasting station is practically perfect even though we realize there is considerable loss of fidelity in most stations.

In the broadcasting studio, sound is translated directly into audio frequency current. This current may be forwarded directly to the transmitter or it may be sent many miles over a telephone line. Along its journey from the microphone to the loudspeaker where it is re-expressed as sound waves, the audio

current is amplified and re-amplified many times. How nearly will the final sound be like the original?

To what extent can a sound be altered before its characteristics are noticeably changed? This leads us to the question, "To what extent can any single frequency in a signal consisting of many frequencies, be amplified or its intensity decreased, before its effect on reproduction is noticeable?" To answer these questions requires exact means of expressing sound and electrical intensities in terms of ear response.

In our present study we are going to assume that the loudspeaker responds faithfully to whatever the audio system supplies it. If the power delivered to the loudspeaker is doubled, the amount of sound energy delivered from it will be doubled and we must assume that the same will be true regardless of the frequency of the audio signal, whether it be 30 or 5,000 cycles per second. In other words, we are going to assume that the loudspeaker does not distort the signal in any way.

Now suppose the loudspeaker is operating at a definite volume (sound level). If the electrical signal is doubled in power, will the ear detect sound twice as loud? Not at all. And we shall see later just how much noticeable increase in sound there will be when the electrical power is doubled. To take a practical example of this—we might have two identical radio receivers, each reproducing programs with equal fidelity, but in the first case the receiver is delivering 1,000 milliwatts of power to the speaker while the other is delivering only 800 milliwatts. The difference in sound intensities from the loudspeakers will not be apparent unless you listen very attentively.

In the broadcasting studio, an orchestra might be playing softly one moment and loudly the next moment. Exact measurements show that the power equivalent of the loud music may be one million times the power equivalent of the soft music. And yet sound engineers say that the difference between the soft and loud music is only 60 decibels, that is, 60 definite increases in sound.

The decibel is not a new word to you. You will remember that it is the smallest change in sound intensity that can be detected by the ear. When the electrical power delivered to a loudspeaker is doubled, as for example from 5 to 10 watts, there is a 3 db. increase in sound output. If the power is increased four times, let us say from 5 to 20 watts, there is a 6 db. increase in sound output. If the power is increased eight times there will

be a 9 db. increase in sound. Suppose we were to increase the power from 1 to 1,000 watts—there would be 30 definite increases in sound output, that is, a 30 db. increase.

The relations given in the preceding paragraph between power in watts and output power in decibels were all worked out from the formula:

$$N_{db} \text{ (number of decibels)} = 10 \log_{10} \frac{P_2}{P_1} \quad (1)$$

Stated in words, the formula says that if the power is increased from P_1 to P_2 , the db. increase is ten times the common logarithm of the power ratio. In Figs. 1 and 2 the relation between watts and decibels is worked out in the form of a scale.*

Returning to the formula, if the power P_1 is increased to P_2 , the sound will be increased N decibels. Suppose the speaker is fed with 100 milliwatts, to what extent will the power have to be increased before the ear can detect an increase of one db. in sound output?

We substitute 1 for N_{db} and 100 for P_1 in formula 1 and we have:

$$1 = 10 \log_{10} \frac{P_2}{100}$$

We divide both sides of this equation by 10 because we want to get the log of the number. We get:

$$\frac{1}{10} = \log_{10} \frac{P_2}{100}$$

From this it is clear that $\frac{P_2}{100}$ is a number whose log is $1/10$ or .1. To find the number whose log is .1 we refer to a log table. We find that the number whose log is 1000 is 1259. As the characteristic of 0.1 is zero, we know that our number is between 1 and 10, so we place the decimal after the first number and get 1.259. Then:

$$\frac{P_2}{100} = 1.259$$

We still have the value of P_2 to calculate. To get P_2 by itself

* If you are unfamiliar with the use of logarithms or wish to review the subject briefly, refer to the last chapter of this lesson, "Logarithms Made Easy."

we must multiply both sides of the equation by 100 and we get:

$$P_2 = 125.9 \text{ mw.}$$

From this we can see that if we start with 100 milliwatts, and we must increase the power to 125.9 mw. before the ear can detect a change, the necessary increase will be 25.9 mw. Or we may say that at the level of 100 milliwatts of power, it requires 25.9 milliwatts to increase the power 1 db.

Suppose the speaker is operating at a level of 10,000 milliwatts. How much must this power be increased before the ear will detect an increase in volume?

$$\frac{P_2}{1000} = 1.259 \text{ or}$$

$$P_2 = 12,590 \text{ mw.}$$

Therefore to obtain 1 db. increase in power at this particular level, an electrical increase of 2,590 milliwatts is required.

By means of the scales in Figs. 1 and 2 which we have worked out for you, you can work out the increase in power required for any db. gain and vice versa, without knowing logarithms. Notice that the upper scale is uniform—it represents db. gain. The lower scale represents power ratio.

The scale in Fig. 1 is not given in great detail as it is merely for the purpose of estimating the approximate db. change for a given power ratio.

Complete detail is shown in Fig. 2 for any single section of Fig. 1. Thus the entire scale in Fig. 2 can be used for close calculations in any of the sections in Fig. 1, marked by the various characteristics, starting at zero and going up to 6.

Now let's see how we are to use these scales. Suppose the power has been increased from 1 milliwatt to 592 milliwatts. In other words, P_1 was 1 milliwatt and P_2 , 592 milliwatts. Then the power ratio between P_1 and P_2 is 592. We want to find out how much the power has been increased in db. We realize that 592 is between 100 and 1,000, therefore we are going to find the db. value in the section which has the characteristic 2 and we are going to expect a db. increase between 20 and 30. For exact calculations we will refer to Fig. 2. On this scale, 592 on the power ratio scale will be just before the division marked 6. Reading on the db. gain scale we see that it corresponds to 772. As we are working in the section of Fig. 1 which has the characteristic of 2, we place a 2 before this last number and we

get 2772. But you will remember we said the db. value would be between 20 and 30. Therefore we place our decimal point after 27 and we have our final result, 27.72 db. gain.

While the use of scales is much simpler than the use of the formula and a table of logarithms, in working out db. gain for a given increase in power, the student is urged to attempt to master the use of simple logarithms. Study the last chapter in this lesson carefully and I'm sure you'll have no difficulty in making any calculations involving simple logarithms.

In using the scales, the power ratio scale can be considered as the power level. Then at 100, in order to obtain an increase of 1 db., that is, from 20 to 21 db., the power ratio must go from 1.00 to 1.26. If the original level was 100 milliwatts, an increase in power to 126 milliwatts would represent an increase of 1 db.

Going back to the problems we worked out by means of the

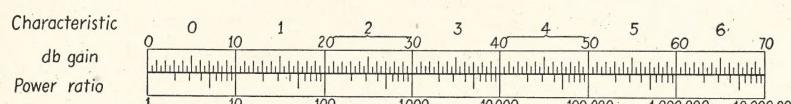


FIG. 1

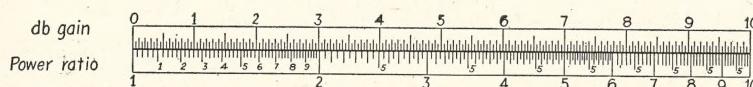


FIG. 2

formula and logarithms, the results of which we can check from our scales in Figs. 1 and 2, we are impressed by the fact that at a level of 10,000 milliwatts, it takes ten times as much change in power to produce an audible change in sound output, as at a level of 100 milliwatts.

Now suppose that a speaker was operating originally at 100 milliwatts and the power fed to it was increased to 10,000 milliwatts. How many perceptible increases in sound output will there be? From our equation No. 1:

$$N_{\text{db}} = 10 \log_{10} * \frac{10,000}{100}$$

$$= 10 \log 100 = 10 \times 2.00 = 20 \text{ db.}$$

Or we can get the same results directly by referring to Fig. 1.

* In the expression \log_{10} , the 10 means that we are using the common logarithm system which is worked out with 10 as the base. As we are always going to use this system, from now on we shall omit the system designation.

Thus there are 20 definite increases in volume although the actual power increase was $10,000 \div 100$ or 100 times.

Volume is increased solely for the purpose of audibility. That is why it is so essential that we have a unit of audibility and that is why the decibel is so important and why it is so necessary to reduce milliwatts to decibels. When there is an increase, we say there has been an amplification or "gain." When there has been a loss in power, we say there is a loss or an "attenuation," sometimes called "negative gain." This too must be measured in db.

Quite often in going through electrical devices, coupling devices, transmission lines, volume controls, etc., a certain amount of power is lost. Thus we have a definite amount of power at the start and less power after it has passed through the device or the transmission line. We talk about power at the source, and power at the *sink*, meaning the output. We often call one end the source and the other end the *load*. If we were to use an amplifier at the sink, which would increase the power to the amount it was originally at the source, we would raise the power a definite number of db. We can also say that the device lost this amplification, or as we commonly say there has been a $-$ db. gain equal to this value.

The attenuation in decibels would be calculated by the same formula as used for gain and in this case the power ratio would be the power output divided by the power input and the answer must be negative db., that is, there should be a minus sign before the db. value.

The formulas for gain and loss in db., in simplified form are:

$$(\text{gain}) \quad \text{db} = 10 \log \frac{\text{larger power}}{\text{smaller power}}$$

$$(\text{loss}) \quad - \text{db} = 10 \log \frac{\text{larger power}}{\text{smaller power}}$$

Under all conditions we must consider the db. value as ten times the log of the power ratio. However, we often deal with voltages and currents in considering gains and losses, in amplifiers and electrical systems. In cases where the voltages or currents are known, it is not necessary to translate them into terms of power for we are also able to express a db. increase or de-

crease by means of voltage ratios and current ratios. It is easy to see why this should be if we remember that power equals $\frac{E^2}{R}$ or I^2R . The formulas to be used in cases where the voltage gain or loss is known are:

$$N_{db} = 20 \log \frac{\text{larger voltage}^*}{\text{smaller voltage}}$$

and the formula to be used in cases where the current gain or loss is known is:

$$N_{db} = 20 \log \frac{\text{larger current}^*}{\text{smaller current}}$$

In other words, the db. gain when using voltages or currents is twenty times the log of the ratio. However, this is exactly true only if the load at the start and the load at the finish both have the same resistance.

REFERENCE LEVELS

When two powers are compared, it is necessary to know the power of each in watts, the current of each or the voltage of each. Then from our formulas or the special scales in Figs. 1 and 2, it is a simple matter to compute the difference between them in db.

In actual practice, all powers are referred to a reference level. For example, you will find that a microphone, a phono-pickup, or a power amplifier is rated as so many db. This means absolutely nothing to anyone unless he knows the power to which it has been compared. In other words, unless he knows the reference or zero db. level.

Unfortunately there are two reference levels in common use which results in some confusion to students and readers of technical articles. For broadcasting, telephone and general radio purposes, the reference level or zero db. is 0.01 watt or as usually stated, 10 milliwatts (mw.). The other reference level or zero db. level used often in sound pictures and public address systems, is 6 milliwatts. However, most engineers consider 10

* Figures 1 and 2 may be used in this case. If we consider the power ratio scale to represent the voltage or current ratio, we must multiply the answer in db. by 2.

mw. as the standard level unless otherwise stated and we shall do likewise.

Now suppose that a dynamic speaker connected to a radio receiver is fed with 3 watts of power. What is the db. level?

The power ratio is $\frac{3}{.010}$ (.010 is our reference level). Dividing 3 by .010 we get 300. From Fig. 1 we find that a power ratio of 300 is equivalent to a db. level of between 20 and 30 and that the characteristic is 2. From Fig. 2 we find that 3.00 is equivalent to 477. Thus the db. value is 24.77.

Taking another practical example, the telephone company states definitely that they will not allow more than 2 db. of power fed into the telephone system. What is the permissible power in milliwatts? From Fig. 2 we find 2 db. is equivalent to a power ratio of 1.58. As the standard is 10 milliwatts, the permissible power is 10×1.58 or 15.8 mw.

One more example: A double button microphone is rated as having a power output of -50 db. What is the power output in microwatts? From Fig. 1, 50 db. equals a power ratio of 100,000. Then if we let X stand for the unknown we have $\frac{10}{X} = 100,000$ and X the unknown level is $\frac{10}{100,000}$ or $\frac{1}{10,000}$ watts. To change to microwatts we multiply by 1,000,000 and we get $\frac{1,000,000}{10,000}$ or 100 microwatts.

In comparing power levels by means of voltages and currents, it is imperative that a standard resistance be used. In other words, the power output and power input must feed into the same value of resistance. For the 10 milliwatt power reference level the load resistance is 600 ohms. We know that P equals I^2R . Then .01 equals $I^2 \times 600$; $I = \sqrt{\frac{.01}{600}} = \sqrt{.0000167}$ or 0.0041 amp. (4.1 ma.).

The voltage across the resistor or the load voltage will be 0.0041×600 or 2.46 volts.

In considering the 6 mw. power level, the resistance load is 500 ohms.

In modern receivers, the load resistance is important, for as in sound transmission, power is always fed into a load having a definite resistance value. A telephone line has a standard effective resistance of 600 ohms at its input and output terminals,

which explains why this value was taken as standard. However, transmission lines may have other resistances and these must always be stated when making current and voltage calculations.

For the purpose of acquainting you with powers, power ratios, db. levels, voltages across and currents through a standard fixed 600 ohm load, the following table which shows a comparison of these factors is included here.

Power in milliwatts	Power ratio to standard	db. level	Voltage	Current (ma.)
.0001	.00001	-50	.00774	.0129
.001	.0001	-40	.0246	.041
.01	.001	-30	.0774	.129
.1	.01	-20	.246	.41
1	.1	-10	.774	1.29
10	1	0	2.46	4.1
100	10	10	7.74	12.9
1000	100	20	24.6	41.
10000	1000	30	77.4	129.
100000	10000	40	246.0	410.
1000000	100000	50	774.00	1290.

A PRACTICAL EXAMPLE OF AUDIO SIGNAL TRANSMISSION

Let us say that a broadcast is being made from a baseball park. A local amplifier takes the audio power from the microphone (double carbon button) and amplifies the signal to the level permitted by the telephone company. The signal is then carried over a telephone line possibly four miles long, without any intermediate amplification, so that a line loss results.

At the studio it is fed into a monitor room where the signal is again raised to the permissible telephone power level. In going to the radio transmitter which let us say is about 20 miles away, a further line loss takes place. At the transmitter the power is raised to zero level before it is fed into the audio modulating system of the transmitter.

The microphone feeds -50 db. of power to the input of amplifier No. 1. This is point 1 in Fig. 3. At point 6, the output of the entire transmission system, the level is to be 0 db. At each point the load resistance is 600 ohms.

Now let us analyze the various power levels, amplification and attenuation throughout the system. At points 1, 2, 4 and 6, the db. levels are -50, 2, 2 and 0, respectively. Let us say that

measurements show the loss in line No. 1 is 7db. and in No. 2, 20 db. Of course, this is the same as saying that the gain in line No. 1 is -7 db. and the gain in line No. 2 is -20 db. These are shown in the diagram.

What are the actual powers in milliwatts at the various points? At point 6 we know that it is 10 mw. for it is 0 db. At point 1 where the power is -50 db., the power level is only .0001 milliwatts. Point 2 is 52 db. above this value and from Fig. 1 we know this corresponds to a power ratio between 100,000 and 1,000,000, the characteristic of which section is 5. This leaves the number 2, which must determine the exact ratio. From Fig. 2 we find the intermediate power ratio corresponding to 2 is 158. Thus the exact power ratio is 158,000. And the level at point 2 is $158,000 \times .0001$ or 15.8 milliwatts.

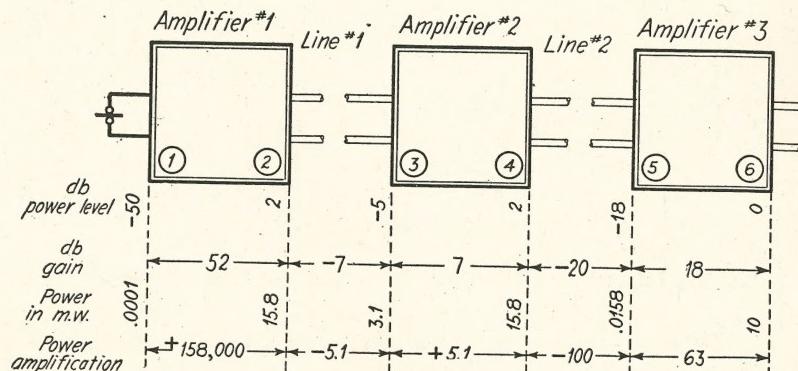


FIG. 3

In going through line 1, 7 db. of power is lost. This corresponds to a power amplification of -5.0 as we find from Fig.

2. The level at point 3 is now $\frac{15.8}{5.0}$ or roughly 3.1 mw. Amplifier No. 2 raises 3.1 to 15.8 mw., to a level of 2 db.

Line No. 2 attenuates the power level of 15.8 mw. by 20 db. From Fig. 1 we find the attenuation is exactly 100 (power ratio)

so the new power level is $\frac{15.8}{100}$ or .0158 mw. Amplifier No. 3 raises the power 18 db. This lies in the section of Fig. 1 having the characteristic 1, that is, the section extending from 10 to 100 on the power ratio scale. From Fig. 2 we find that the power

ratio corresponding to 8 db. is 6.3. Thus the actual amplification is 63 times.

As a final check we can multiply 63 by .0158, which will give us 9.95, which is very close to 10 milliwatts, close enough for rough calculation.

The chart in Fig. 4 will enable you to visualize sound intensities at various db. levels. At the lower left the zero sound level is just at the point of audibility, the smallest sound that the ear will detect. This chart refers only to actual sound energy in bars.* The figures given are only true for a single frequency

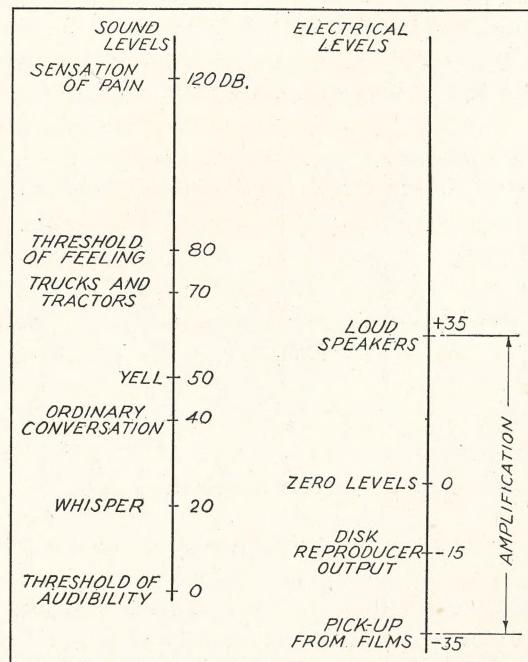


FIG. 4

and a 1,000 cycle note is usually the reference standard. This particular line illustration shows that if the sound is increased 20 db., we shall have a whisper; 40 db. is ordinary conversation; 50 db. a yell, and between 80 and 100 db., the sound is increased just to the point before it is actually *felt* and becomes painful.

* A cubic centimeter of water would exert a downward force of one dyne due to gravity. If a force of one dyne is exerted on a surface of one square centimeter, a pressure of one "bar" exists. At 1000 cycles the threshold of audibility to the average ear is .00052 bar.

This sensation of pain is no doubt familiar to you. When you listen to an exceptionally loud sound, at low frequencies, you actually feel and are disturbed by the vibration. On the other hand if the sound has an extremely high frequency, it will cause a ringing in your ears and if it continues long enough it may result in a severe headache.

The range of audibility from the threshold of audibility to the point of feeling is about 80 db. Common sound levels range from whispering (20 db.) to ordinary conversation (40 db.) and even to the point of shouting which is 50 db. In ordinary sound picture work, a range of 30 db. (from 20 to 50) is large enough for practically all requirements.

To the right of the figure is the line representing the electrical reproducing end of sound pictures and the required electrical power input to the loudspeaker measured in db. It is interesting to know that zero db. (10 milliwatts of power) is the power output of an ordinary desk telephone transmitter. -15 indicates the number of db. below this level, representing the power at the pickup output. Notice that the output from the pickup using a film as the record is 35 db. below the zero level. For loudspeaker operation it is necessary that the signal be increased by means of amplifiers 35 db. above zero level or a total of 70 db. from the time it leaves the film to the point where it is thrown out to the audience or into an auditorium.

AUDIO RESPONSE CURVES

The loudspeaker is the final load in a radio receiver. To it must be fed an audio current, an electrical duplicate of the original sound. An electric motor is a close analogy, where mechanical power is obtained from electrical power. The speaker goes one step farther however, it also converts mechanical energy into sound energy.

To be exact, the amount of sound energy delivered by the moving diaphragm of a speaker is quite small, for the efficiency of the best speakers is rarely over 7 per cent and the average is nearer 2 per cent. If 1,000 milliwatts of electrical power are supplied to the speaker input, only 20 to 70 milliwatts are actually available as sound energy. A very low efficiency to be sure, but a condition which radio men must contend with. Today every effort is being made to increase speaker efficiency.

The power output of an audio system is usually considered

as the power delivered to the input of the speaker unit. For the ordinary home a power output of 1 to 3 watts (20 to 24.7 db.) is considered sufficient. The load of a speaker may be reasonably considered as a resistance load connected directly in the plate circuit whose power loss is the power output of the audio system. For example, if the power output is P , the effective resistance of the speaker is R and the voltage across the speaker is E_o , the power fed to the speaker is found from the following equation:

$$P = \frac{E_o^2}{R} \quad \text{where } P \text{ is in watts}$$

E_o is in volts
 R is in ohms

We usually say that a speaker will handle a definite number of watts faithfully, that is, without any appreciable distortion. A speaker may have any value of resistance and by means of an appropriate ratio audio transformer (usually called an

Type	Maximum Undistorted Power Output Milli-watts	Maximum Grid Voltage	Maximum Plate Voltage	Plate Resistance In Ohms	Required Plate Load In Ohms
'120	110	22.5	135	6,300	6,500
'131	150	22.5	135	4,950	9,000
'112A	260	13.5	180	5,000	10,800
'171A	700	40.5	180	1,950	3,900
'245	1,600	50.0	250	1,750	3,900
'250	4,600	84.0	450	1,800	4,350
Pentode	2,500	16.5	250	35,000	8,375

FIG. 5

impedance matching transformer), the effective resistance of the speaker can be raised or reduced to any reasonable value. This will be taken up in a later text. Under these conditions we are most interested in the voltage required to operate the speaker, which is found from the following formula:

$$E_o = \sqrt{P \times R}$$

In spite of the fact that the audio output is governed by the ability of the speaker to handle power, it generally is assumed that the output of the power tubes is the criterion for power output of the audio system. The speaker then is made sufficiently large to handle what is delivered to it. Therefore, in considering the sound output, consider the maximum output of the tubes used.

As you already know, there are two maximum outputs of a tube—the maximum power output and the maximum undistorted power output. Under all conditions the maximum power output is obtained when the plate resistance of the tube is equal to the impedance of the speaker. This may be realized by using a proper matching transformer. The maximum undistorted power is obtained when the load impedance is between 1 and 2.5 times that of the tube.* Here we must be guided by the recommendations of the tube manufacturer or by an analysis of a family of tube curves.

Referring to the table given in Fig. 5, if a '50 tube is used in the output stage whose plate resistance is 1800 ohms, the load resistance for maximum undistorted power of 4.05 watts is

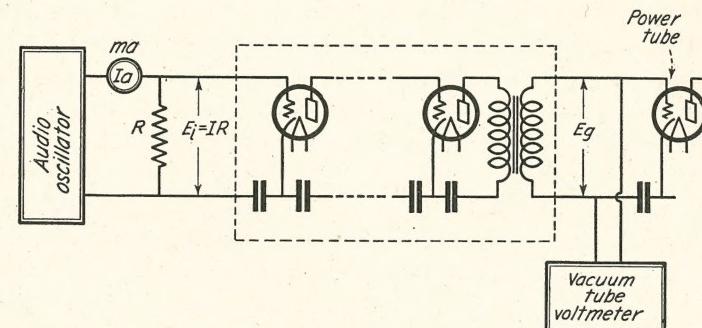


FIG. 6

4350 ohms. This is according to the recommendation of tube manufacturers. Suppose, then, the speaker by means of a matching transformer was made equal to this value. What voltage would be necessary to drive 4.05 watts into it? From our formula:

$$E_o = \sqrt{4.05 \times 4350} = \sqrt{17620}$$

$$= 133 \text{ volts}$$

Knowing the amount of voltage available at the input of the audio system, we can immediately compute the required voltage amplification of the entire audio amplifier. This is referred to as the over-all voltage gain (G_v). If the input voltage is E_L , then the required voltage gain is

$$G_v = \frac{E_o}{E_L}$$

* An exception to this is in the case of the Pentode tube. To obtain maximum undistorted power output from a Pentode, the load impedance should be from 1/5 to 1/4 the internal resistance of the tube.

The efficiency of the audio amplifier is determined by the gain of the system as a whole and by how much deviation from this gain there is as the frequency is varied. The gain of an audio amplifier and its frequency response are measured by means of an audio oscillator and indicating device, as shown in Fig. 6. In this case the voltage input to the power tube is measured. Exact methods for making gain measurements are given in the advanced Radio Servicing Course.

Voltage amplification measurements should be taken over a frequency range from 30-10,000 cycles and plotted in the form of a curve, as shown in Fig. 8.

This curve will now give the audio characteristic of the audio amplifier and will show how well the low and high fre-

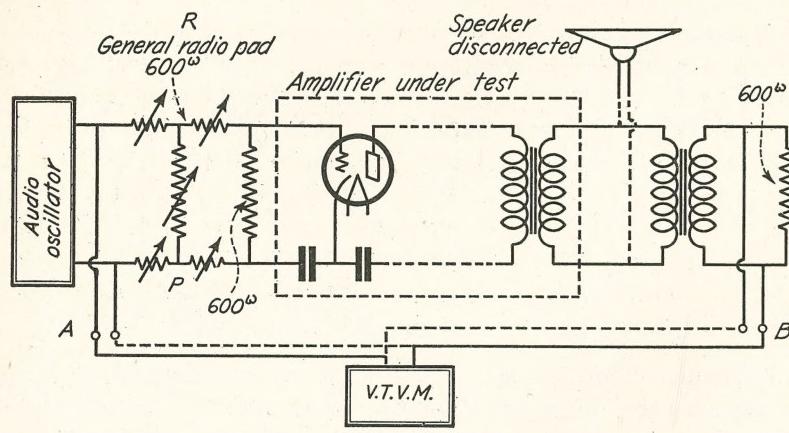


FIG. 7

quencies are amplified as compared to the frequencies which are always easy to amplify (400-1,000 cycles). For best reproduction in the speaker, this characteristic curve should be as flat as possible, unless some undesirable characteristic of the power tube or speaker requires a different audio characteristic.

Voltage amplification curves convey considerable information, but when we wish to take into consideration every possible distortion or loss in an audio system it is best to consider the input and output powers in terms of db. By feeding a voltage into an amplifier across a definite value of resistance and by using an output load of equal ohmic value, if the voltage input is E_1 and the load voltage is E_2 , then the db. power level is:

$$N = 20 \log \frac{E_2}{E_1}$$

By assuming that the db. level at a certain frequency is standard and measuring the db. increase or decrease at other frequencies a good indication of the faithfulness of the audio system may be obtained.

A method suitable for measuring the true over-all audio decibel gain is shown in Fig. 7.

Pad P is a 600 ohm constant impedance attenuation unit, calibrated in db. The indicating device V.T.V.M. is a volume indicator, which is nothing more than a tube voltmeter, arranged so that it may be connected to the input or the output. The secondary of the output transformer is coupled to a 600 ohm resistor through a matching transformer, so that the 600 ohm are reduced to an impedance equal to that of the voice coil of the dynamic speaker used with the amplifier. In making the measurements, for a given frequency the volume indicator V.T.V.M. is first placed across the input A and the audio oscillator adjusted for a suitable meter reading. Without readjusting the oscillator the volume indicator is placed across the output B and the pad P is adjusted until the same deflection is noted on the output meter. This is generally done first at 400 cycles, which is the reference level frequency. The same procedure is followed over the complete frequency range and the results plotted. The following table and Fig. 9 will make this clearer.

Frequency	DB on pad P	DB Deviation from 400~
30	35	-9
50	40	-4
100	42	-2
200	44	0
400	44	0
1,000	44	0
2,000	43	-1
4,000	42	-2
6,000	40	-4
10,000	37	-7

In Fig. 9, the complete over-all audio characteristic is shown. This is known as the complete "audio-response" of an audio amplifier. From this curve one would say that at 30 cycles the gain of the audio amplifier is -9 db. down, having in mind the reference level of 400 cycles, which has become more or less standardized.

POSSIBLE SOURCES OF AUDIO DISTORTION

If every frequency from 30 to 10,000 cycles were amplified uniformly by the audio system, that is, from input to output, there would be absolutely perfect fidelity for practically every audio purpose. Each frequency with its harmonics would be unaltered, hence no distortion would take place. But gain and fidelity curves show that this is only approached in actual practice. What factors enter into audio amplifier construction which prevent perfect undistorted amplification? Let us consider the major sources of distortion.

Permissible Grid Swing. The expression "grid swing" has been used often, but the permissible swing has not been studied in detail. Fig. 10 represents the E_g - I_p curve of either an intermediate or output tube. The operating point is at 1, set by the value of the grid bias. An A.C. input voltage, whose effec-

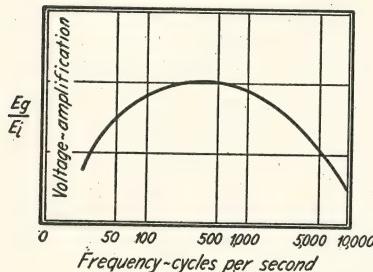


FIG. 8

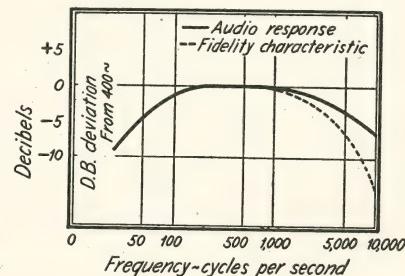


FIG. 9

tive value is E_g (r.m.s.), is fed to the grid. When power is considered, r.m.s. values only can be used. The amplitudes (peak voltages) must be considered when estimating the grid swing. Peak voltages for a sine wave voltage are determined by the formula:

$$E_{\text{peak}} = 1.41 \times E_{\text{r.m.s.}}$$

The total grid swing is twice this or S, as in Fig. 10, that is:

$$S = 2.48 \times E_{\text{r.m.s.}}$$

In this particular E_g - I_p curve, the grid voltage swings over the straight portion of the curve. As long as the swing is not greater than twice 1 to 2 or from 2 to 3, all is well. As soon as the swing is beyond 2, the grid at the positive maximum grid swing becomes positive and the grid acts like a miniature plate, drawing a grid current, by attracting electrons. We must not

overlook the fact that the grid voltage is supplied by a device which has an internal impedance and consequently its terminal voltage will drop as soon as current flows in the grid circuit. The impressed voltage, instead of following the actual wave form, drops at this point and the output current form is no longer like the original, but is cut off at the top (shown shaded). The result is false harmonics, and distortion. The remedy is to increase the C bias which will be satisfactory provided the grid swing is within the straight portion of the E_g - I_p curve.

We know that audio signal currents can be said to consist of one or more fundamentals and their harmonics. Each by itself is a pure sine wave and each voltage wave must pass through the tube without alteration of its form. Any action which causes the positive part of the cycle (of an isolated frequency) to differ in wave form from the negative part introduces undesirable harmonics and consequently, distortion. A D.C. milliammeter placed in the plate circuit will show a change in D.C. plate current, and will show whether distortion is present. When the grid swing is such that the grid itself becomes positive, the average plate current is lowered and the needle will dip toward zero. We have learned that the negative C bias must be great enough to prevent a positive grid and this condition is realized when the half grid swing is equal or less than the C bias.

Even if the C bias is of the proper value to prevent the grid from swinging positive, the wave form of the plate current may still differ from the grid voltage wave form. Fig. 11 shows a possible E_g - I_p curve with the C bias E_1 of such a value that the point of operation is at 1. E_g is the grid voltage wave form and is perfectly symmetrical (sine wave). Note that the plate current is determined by the swing over the E_g - I_p curve in this case from 3-1-2. But below 1 the curve has a decided bend in it. The result is a distorted I_p and the ear is aware of distortion.

A "C" bias bringing the operating point up to 2 would cause the tube to operate on the straight portion of its curve. The grid should not swing past E_1 and a, or a positive grid will cause operation on the curved portion and will introduce distortion. The remedy, if a large grid swing is essential, is to increase the plate voltage and the C bias or to increase the load resistance. In this case, a milliammeter in the plate circuit would show a rise in plate current.

The effect of increasing the resistance in the plate circuit should be understood. Although Fig. 10 and Fig. 11 are static E_g - I_p curves (no load resistance), we must consider the dynamic E_g - I_p characteristic for a true understanding of conditions. A large load (greater impedance) results in a flattening out of the E_g - I_p curve, but this advantage is offset to some degree by the loss of output power when the tube is an output tube. In the case of a voltage amplifier there is a decided gain.

Inspect Fig. 12a very carefully. E_g is the grid input voltage. The voltage between 1 and 2 is the bias, and the voltage between 2 and 3 is the plate supply voltage. These last two

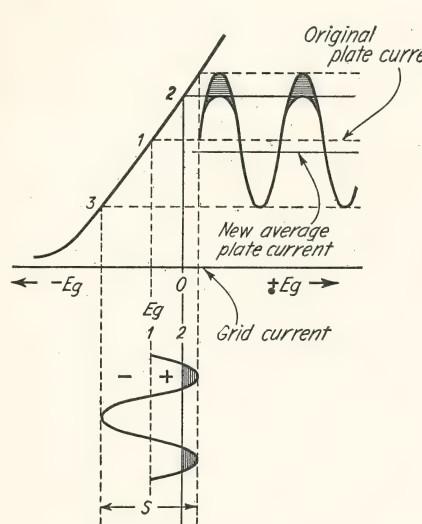


FIG. 10

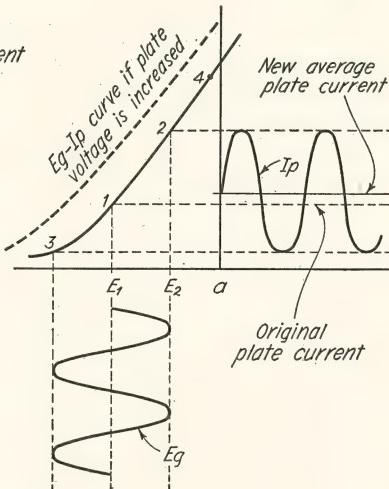


FIG. 11

determine the plate current, represented by the operating point a in Fig. 12b for a tube operated with a plate voltage of 180 volts. This is the voltage between 2 and 4 in Fig. 12a. Let us say that E_g is rising (it is an A.C. potential) becoming more positive. Naturally this makes the net grid voltage less and the operating point we would say is now at b in Fig. 12b. But let us think well before we accept this.

The voltage V_{2-4} —that is, the voltage between 2 and 4—together with the net grid voltage determines the plate current. The assumption that the curve ab applies to all conditions rests on the fact that the voltage V_{2-4} does not change. However, the supply voltage V_{2-3} less the IR drop (V_{3-4}) is the determining

plate voltage. V_{2-3} , the plate supply voltage, always remains the same in operation (dynamic conditions); the IR drop will increase with increased plate current and decrease with decreased plate current. Naturally, increased plate current will decrease V_{2-4} and decreased plate current will increase V_{2-4} and the original assumption that V_{2-4} is constant is not true.

Going back to the original conditions, as the net E_g is made less negative, the current goes up, the IR drop increases and V_{2-4} becomes less with the result that the actual plate current is b_1 instead of b —see Fig. 12b. On the other hand, if the signal voltage swings negative, the IR drop becomes less and the actual current is C_1 instead of C . The entire effect is to flatten out the E_g - I_p curve and the amount of flattening out depends on the load resistance R . Fig. 13 shows a typical condition with various load resistances. Clearly, the larger the resistance

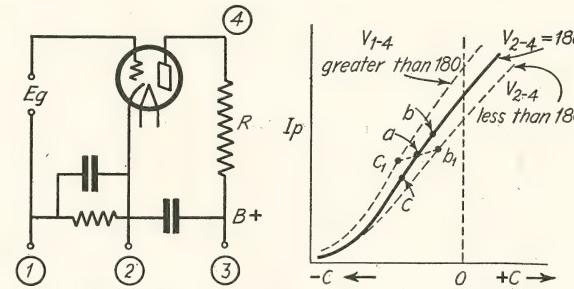


FIG. 12a

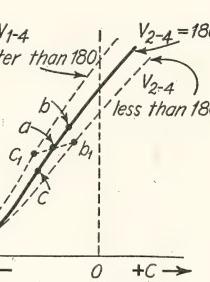


FIG. 12b

the flatter the E_g - I_p curve and the less distortion due to E_g - I_p curvature. Of course, in output tubes, too large a load resistance would decrease the power output, hence a value is used to obtain maximum undistorted power output.

A. F. TRANSFORMERS

Transformers have a double duty to perform in audio amplification stages. First, its primary impedance must be sufficiently high to have impressed across it a large voltage. The grid voltage E_g is amplified by the tube μ times, and the voltage in the plate is μE_g . This voltage divides between the plate impedance R_p and the load impedance Z_p (primary). The larger voltage is naturally that across the larger impedance. R_p is resistive while Z_p is inductive and if the latter is made large enough, it may have practically the full voltage μE_g across it. The second duty is to increase the voltage in the secondary by transformer step-up.

The plate impedance R_p is a governing factor in audio step-up ratios. A detector tube is operated at low voltage, and a tube chart will reveal that its plate impedance is high at low plate voltages. It, therefore, should have an impedance load of much greater value than the second audio tube which may also have a transformer as the tube will be operated at a higher plate voltage. A larger number of turns on the primary of the first audio transformer is required, and for economy's sake the turn ratio of the transformer is kept low in order to limit the number of secondary turns. Even if a large number of turns are used the secondary may have so much distributed capacity that it, with the secondary inductance, may be at resonance in the audio range. Fig. 14 shows this clearly. A represents a

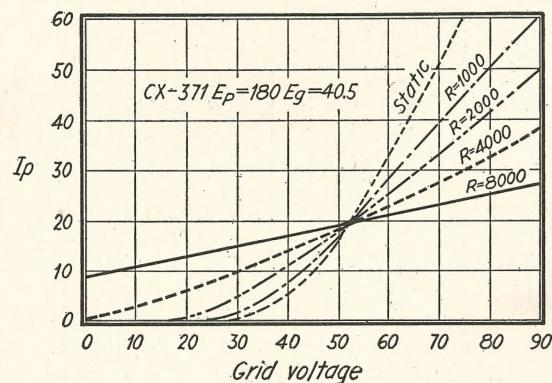


FIG. 13

low ratio transformer and *B* a high ratio transformer, both of conventional construction. Note the large hump in the curve of the high ratio transformer indicating a resonance peak. The general practice today, when using a two-stage transformer coupled amplifier, is to use a low ratio transformer in the first stage (2-1, 3-1) and a second stage transformer of 4-1 or 6-1.

Distortion in audio transformers is an important item, and the serviceman should know what is done to eliminate it as far as possible. In radio the push-pull audio operates with current in the primary in a split feed manner which cancels the magnetizing flux in the core.* In contrast with this, transformers used in single audio stages have the plate current flowing

*This is brought out in greater detail in a later chapter on push-pull amplification.

through their primaries with the result that the cores always carry varying flux densities, when the amplifier is in operation.

Fig. 15 shows the magnetization curve of a magnetic circuit. At low currents the flux (ϕ) rises slowly as the current increases as shown by *a-a*; from *a* to *b*, the increase is proportionate (straight characteristics); from *b* the ϕ rises very little for even great changes in current (plate current). All this we have studied before. With a push-pull transformer the only effective current in the primary is the A.C. signal current, and the actual change in flux will follow hysteresis loops *a-a'*, *b-b'*, *c-c'*, etc., as shown in Fig. 16. The incline of the line drawn from *a-a'*, *b-b'*, etc., is a measure of the permeability of the magnetic circuit. The more vertical the line, the greater is the permea-

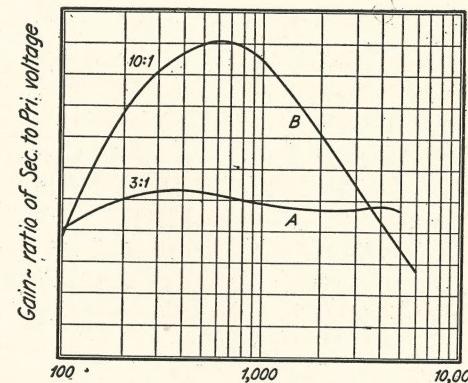


FIG. 14

bility. As you know, the inductance of the primary will be increased directly as this permeability is increased.

With small signal current the loops will be small (*a-a'*, *b-b'*) and the permeability will be low. Further increase in signal will cause increased permeability as will be clearly seen by analyzing the slope of *b* to *b'*, and *d* to *d'*. A swing beyond *d* will again reduce the permeability and the primary inductance as shown by *c* to *c'*. As we must have as much primary inductance as possible and with little change, the primary turns and the magnetic circuit must be so designed that the signal swing will be from *d* to *d'* as in Fig 16. Clearly a primary with many turns of wire will require a large magnetic core.

The secondary voltage wave form depends on the slope of the curve and will follow the hysteresis loop when only the sig-

nal current is present. Distortion is bound to occur. To overcome this and at the same time make the magnetization curve of Fig. 15 straighter and the loops in Fig. 16 thinner and almost identical in shape, an air gap is introduced in the magnetic circuit. The result is constant inductance and decreased wave distortion.

In an ordinary audio stage, a D.C. current does flow in the primary, and thus the magnetizing current never flows in the opposite direction. When an A.C. signal voltage is added, the flux changes can be represented by small loops, whose slope will be less than the original magnetization curve. Consequently, the primary will have less inductance than if the same primary were used for push-pull. The slope will be uniform for equal signal swings if the D.C. current operates the magnetic current over the straight portion of the magnetization curve, Fig. 15.

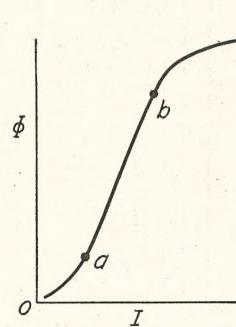


FIG. 15

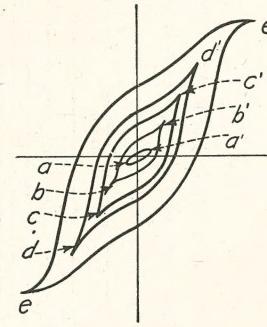


FIG. 16

Operating below a and above b is to be avoided. Again hysteresis will distort the wave form unless sufficient air gap is used in the magnetic circuit.

PUSH-PULL AMPLIFICATION

The purpose of the push-pull amplifier is to eliminate, as far as possible, distortion caused by tube overloading. It employs two tubes to do the work of one—each tube doing half the work—and by dividing the load in a special arrangement, each does its job efficiently; and the sum total of their efforts is a clear, rich audio signal for the loudspeaker.

Let us go back for a moment and consider the results of tube overloading. Figure 11 shows a characteristic amplification curve, on which we assume a point midway between 1 and 4 as the operating point. Ideal amplification is obtained when any increase in grid voltage E_g is marked by a proportionate increase in plate current I_p . Thus, as long as the voltage swing E_g does not extend beyond 1, the straightest portion of the E_g - I_p curve, the wave form of the plate current will be undistorted. But suppose a large incoming signal causes the grid voltage to swing from 3 to 4. Then the plate current

will not increase proportionately between 3 and 1. The result will be a cramped wave form with the lower peaks flattened out.

This distortion of the wave form has been very carefully studied and it has been shown that it consists of the true wave form plus other wave forms which are two, three, four, five times, etc., the frequency of the ideal or fundamental wave. They are called distortion harmonics.

Of these frequencies in the distorted wave form, the second is the trouble maker. The second harmonic current is the most prominent, with the third harmonic next to contend with.

Distortion due to tube overloading is not so apt to occur in stages preceding the last, because grid swing voltages are comparatively small. But the last stage is frequently called upon to handle a large grid voltage swing. Of course, there are single power tubes that can be used, as for instance, the '50 tube or the '10 tube which can handle a large grid voltage swing, but they are costly and require a special, high voltage B supply. Right here is where the push-pull arrangement becomes valuable—and the push-pull amplifier not only reduces the danger of overloading by dividing the load, but actually permits a considerable degree of overloading because the worst of the harmonics, the second, and its related harmonics (the fourth, sixth, eighth), are *actually balanced out* of the circuits. So tubes in a push-

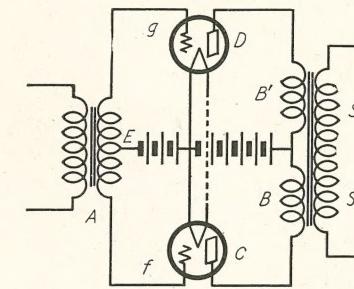


FIG. 17

pull arrangement can be overloaded to a point where the third harmonic becomes objectionable, without a trace of distortion in the speaker. This arrangement then gives us the fundamental or real wave with so little of the distorting third harmonic that its presence is not noticeable in the loudspeaker.

Before we go on to see how this is accomplished, let us see just what a stage of push-pull amplification amounts to. Figure 17 shows the arrangement schematically. C and D are the two tubes. They must be of identical characteristics—that is, their E_g - I_p curves must be exactly the same. The input transformer, A , has its secondary winding tapped exactly in the center. The center tap E , is for the "C" connection (either a "C" battery or C bias resistance in A.C. receivers). When a voltage is induced in the secondary of A , half acts on the grid of tube D and half on the grid of tube C (E is now the grid return). But when the grid of D is positive, the grid of C will be negative. Accordingly we say that the two grids are excited in phase opposition—or that the grid voltage change in D is 180° out of phase with respect to the grid voltage of C .

Suppose a grid bias of 35 volts is required for the operation of either tube over the center of the straightest portion of its E_g - I_p curve. (This means that grids f and g are both at a potential of 35 volts with respect to the filaments.) If a peak A.C. voltage of 30 (the audio signal) appears

across the secondary winding of the input transformer, *A*, the grid voltage of one tube will be increased by 15 volts, while the grid voltage of the second tube will be reduced by 15 volts. The incoming voltage is then divided between the two tubes, and the individual tubes, instead of swinging from 35 plus 30 to 35 minus 30, swing from 35 plus 15 to 35 minus 15, which would possibly be well within the straight portion of their amplification ($E_g - I_p$) curves.

Now we can get back to the interesting problem already mentioned—balancing out the distortion which may appear when the grid swing is large. What happens can be stated very simply—because the grids are fed in opposite phase, the plate currents will be in opposite phase throughout a voltage change in *A*. The second harmonic, however, has a frequency double that of the fundamental so that this second harmonic will be $2 \times 180^\circ$ or 360° out of phase, which is exactly in phase. The even harmonic currents generated in one tube are made to buck the corresponding currents generated in the other tube, and kill each other, when they meet in the primary of the output transformer.

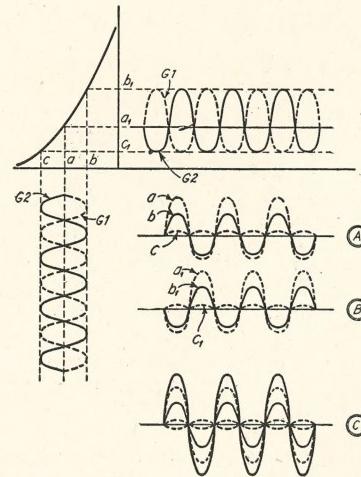


FIG. 18

Fig. 18 shows a distorted current wave form *G*₂ resulting from a tube operating on the curved portion of its characteristic. Notice the flattening effect at the bottom of the curve because an increase in grid voltage *ab* and *ac* is not marked by a proportionate change in plate current. Fig. 18A shows this same curve resolved into its component parts: "a" is the distorted curve, "b" is the fundamental curve, or what it should be if conditions in the tube were ideal, and "c" is the second harmonic. Note that it is twice the frequency of the fundamental. The second harmonic plus the true wave is the distorted wave.

Fig. 18B shows what is going on in the other tube at the same instant. Wave curves *A* and *B* are opposite—or 180° out of phase. When we subtract the fundamental *b*, from the distorted wave curve *a*₁, we get *c*₁ which is in phase with *c*, the second harmonic in the first tube.

The distorted plate currents are now led into the ends of a double primary output transformer with the primaries connected together at the center and acting on the same core. They are arranged so the currents which are in phase create magnetic fields which wipe each other out, then all currents

which are out of phase will build magnetic fields on each other and pass on to the secondary and loudspeaker.

In Fig. 18C we see what happens to the wave forms in the output transformer. Let us take the plate current in tube *C* as represented in 18A and the plate current in tube *D* as represented in 18B and lead them into coils *B* and *B'* respectively (see Fig. 17). Remember that whatever the net results of the current in the primary (*B* and *B'*) are they are reproduced by induction in the secondary *S*. We want to retain the fundamental wave so the plate currents of *C* and *D* are led in, in such a way that they—the fundamentals—add, that is, they are brought into phase. In doing this we throw the second harmonics, which were in phase, out of phase, and, being identical in value they cancel out, ridding the circuit of the distortion. 18C is the result; the harmonics subtract while the fundamentals add.

As far as third and odd harmonics are concerned, they will not be effective until the tube is considerably overloaded, because even as much as a 5 per cent distortion will almost be unnoticeable in the loudspeaker.

LOGARITHMS MADE EASY

What do we mean by a "logarithm"? We know that $10 \times 10 = 100$ and that we may express 100 as 10^2 . Likewise, 1,000 may be expressed as 10^3 ; 10,000 as 10^4 ; and 100,000 as 10^5 . The expression 100, 1,000 or 10,000 is referred to as a "number." The expression 10^2 , is made up of two parts, 10 being the base and the figure 2 is called the *power*, the *index*, or the *exponent*.

Let us multiply $10^2 \times 10^3$. Thus—

$$10^2 \times 10^3 = 10 \times 10 \times 10 \times 10 = 10^{2+3} = 10^5$$

In other words, if we want to multiply any two numbers—a multiple of 10—we merely add the powers. The important fact for us to know is that any number—2, 3, 4, 5, 27 or 923—may be expressed as 10 with an exponent, and this exponent is called the "logarithm." Thus—

$$2 = 10^{-301}; 3 = 10^{-477}; 4 = 10^{-602}; 5 = 10^{-699}; 27 = 10^{1.431} \text{ and } 923 = 10^{2.96}$$

However, the common practice is to express the facts in this manner; "the logarithm of 2 is .301" or in mathematical shorthand:

$$\log_{10} 2 = 0.301; \log_{10} 923 = 2.965, \text{ etc.}$$

But bear in mind that this is only another way of expressing the above. Every number has a logarithm and every logarithm represents a number. Mathematicians are able to calculate the logarithm of any number but for us it is sufficient to have a logarithmic table available and when we are interested in obtaining the logarithm of a number, we merely refer to this table.

You will notice that every logarithm is made up of two parts, separated by a decimal point. The part to the right is known as the "mantissa" and the part to the left as the "characteristic." For example, the log of 923 (2.965) is made of two parts, .965 the mantissa, and 2 the characteristic. Logarithmic tables only give the mantissa. Consequently, if you are dealing with a 3 place logarithmic table, you merely consider the first three numbers in referring to the log table. Thus with 1725, 18.72, .0927. Look up in the log table the mantissas of 172, 187 and 927.

The characteristic is determined by the size of the number under question. The characteristic for numbers from 1 to 10, exclusive, is zero. From 10 to 100, 1; from 100 to 1,000, 2, etc. Thus the characteristic of 27 is 1, the mantissa is .4314. The logarithm is 1.4314.

A table of logarithms is essential if any db. computations are contem-

plated. One is given elsewhere in this course but a more complete table of logarithms may be purchased for a nominal sum.

What is the rule for obtaining the logarithm of a number if the number is less than 1? In the same manner, without regard to the decimal point, determine the mantissa from the log table. The characteristic will be as follows: from .1 to 1 will be -1 ($\log_{10} 9 = -1.954$); from .01 to .1, the characteristic will be -2; from .001 to .01, the characteristic will be -3 ($\log_{10} 0.0097 = -3.987$). The characteristics, in general, can be obtained by comparing the number to the following table of numbers in multiples of 10.

$10^3 - 10^4$	1000 - 10,000 =	3.---
$10^2 - 10^3$	100 - 1,000 =	2.---
$10^1 - 10^2$	10 - 100 =	1.---
$10^0 - 10^1$	1 - 10 =	0.---
$10^{-1} - 10^0$.1 - 1 =	-1.---
$10^{-2} - 10^{-1}$.01 - .1 =	-2.---
$10^{-3} - 10^{-2}$.001 - .01 =	-3.---
$10^{-4} - 10^{-3}$.0001 - .001 =	-4.---

To multiply two numbers, three numbers or four numbers, merely find the logarithm of each number which includes, of course, the characteristic and its mantissa. Add the logarithms together and from the log table determine the number which this logarithm represents.

To divide one number by another, find the logarithm of each number and subtract the second one from the first. The remaining logarithm represents the number and this may be found by referring to the log table. For example: we multiply 59.7×62.4 thus:

$$\begin{array}{rcl} \log 59.7 & = 1.7760 \\ \log 62.4 & = 1.7952 \\ \hline \log \text{Number} & = 3.5712 \end{array}$$

Number = 3726 — that is the number whose log is 3.5712.

Now let us divide 976 by 42.1.

$$\begin{array}{rcl} \log 976 & = 2.9894 \\ \log 42.1 & = 1.6243 \\ \hline \log \text{Number} & = 1.3651 \\ \text{Number} & = 23.2 \end{array}$$

No difficulty will be experienced in division if the log which is subtracted is less than the log from which it is to be subtracted. Here is a case, however, which may entail some difficulty; $59.4 \div 62.4$:

$$\begin{array}{l} (1) \log 59.7 = 1.7760 \\ (2) \log 62.4 = 1.7952 \end{array}$$

Quite clearly, 2 cannot be subtracted from 1 as it stands but a very simple trick will permit us to carry through this process. Consider 59.7 as 597, that is, increase it ten times. When the final answer is obtained, divide the result by 10 and you will have the correct solution. Therefore, our problem is reduced to $597 \div 62.4$ and we solve as follows:

$$\begin{array}{rcl} \log 597 & = 2.7760 \\ \log 62.4 & = 1.7952 \\ \hline \log \text{Number} & = .9808 \\ \text{Number} & = 9.56 \end{array}$$

Then, dividing by 10, we get .956 and the problem is solved.